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EXAMINER	
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Please find below and/or attached an Office communication concerning this application or proceeding.

The time period for reply, if any, is set in the attached communication.

Office Action Summary	**Application No.**	**Applicant(s)**	
	10/532,593	WRAY ET AL.	
	Examiner	**Art Unit**	
	Christopher Crutchfield	2466	

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE *3* MONTH(S) OR THIRTY (30) DAYS, WHICHEVER IS LONGER, FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on *14 February 2011*.

2a)☒ This action is **FINAL**. 2b)☐ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) *1,2,7-9 and 14-19* is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) *1,2,7-9 and 14-19* is/are rejected.

7)☐ Claim(s) _____ is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☒ Notice of References Cited (PTO-892)

2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)

3) ☐ Information Disclosure Statement(s) (PTO/SB/08) Paper No(s)/Mail Date _____.

4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .

5) ☐ Notice of Informal Patent Application

6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

1. The following is a quotation of 35 U.S.C. 103(a) which forms the basis for all obviousness rejections set forth in this Office action:

> (a) A patent may not be obtained though the invention is not identically disclosed or described as set forth in section 102 of this title, if the differences between the subject matter sought to be patented and the prior art are such that the subject matter as a whole would have been obvious at the time the invention was made to a person having ordinary skill in the art to which said subject matter pertains. Patentability shall not be negatived by the manner in which the invention was made.

2. The factual inquiries set forth in *Graham* v. *John Deere Co.*, 383 U.S. 1, 148 USPQ 459 (1966), that are applied for establishing a background for determining obviousness under 35 U.S.C. 103(a) are summarized as follows:

> 1. Determining the scope and contents of the prior art.
> 2. Ascertaining the differences between the prior art and the claims at issue.
> 3. Resolving the level of ordinary skill in the pertinent art.
> 4. Considering objective evidence present in the application indicating obviousness or nonobviousness.

3. This application currently names joint inventors. In considering patentability of the claims under 35 U.S.C. 103(a), the examiner presumes that the subject matter of the various claims was commonly owned at the time any inventions covered therein were made absent any evidence to the contrary. Applicant is advised of the obligation under 37 CFR 1.56 to point out the inventor and invention dates of each claim that was not commonly owned at the time a later invention was made in order for the examiner to consider the applicability of 35 U.S.C. 103(c) and potential 35 U.S.C. 102(e), (f) or (g) prior art under 35 U.S.C. 103(a).

4. **Claims 1, 2 and 16** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application

Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1)

Regarding claim 1, *Qiu* discloses a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising:

a. Determining a packet loss rate previous transmissions to a network (Pages 872-874, Particularly Sections 2-4). (The system of *Qiu* discloses a call admission control system that performs measurement based admission control [Page 872, Section 2]. The system operates by sending continuous probes to a second network endpoint in order to determine network delay and loss characteristics [Page 873, Section 2, Left Column, First Full Paragraph]. The loss and delay characteristics are then used to determine the delay shape distributions by tracking the probability of various packet delays for various packet loss rates [Page 873, Section 3]. For example, when the desired loss rate [i.e. state "0"] is set to 1%, the system tracks the "delay shape" [i.e. the average delay- the minimum delay] for all measurements with a loss rate less then the “0” state [i.e. all measurements with a loss rate <1%] and for all measurements with a loss state of “1” [i.e. all measurements where the loss rate exceeded 1%] [Page 873, Section 3, Particularly Fig. 2 and the Preceding Paragraph]. The intersection of the delay shape functions for the "0" and “1” state is then used to determine the low and high delay shape distribution profiles [Page 873, Section 3, Particularly Fig. 3][See also Page 873, Page 4, Particularly Equations (2) and (3) - Showing the calculation of the low and high delay shape profiles]. Whenever the system detects an incoming connection control

request, it compares the delay state from the last set of network probe packets with the delay state profiles [which are based on past packet loss rate and delay - i.e. the “packet loss rate of previous network transmissions”] [Pages 872-873, Section 2, Particularly Fig. 1]. If the delay shape is below the high delay shape profile but above the loss delay state profile, then the system will send a test probe to the network to determine the current network loss characteristics [i.e. the “current packet loss rate”] and will admit or reject the transmission based on if the current loss rate is above or below and admission threshold [i.e. “decide to drop an attempt based on the current packet loss rate”] [Fig. 1, and Page 872, Last Paragraph, Carried on to Page 873][See also Page 873, Last Paragraph, Carried onto Page 874]. If the delay shape is above the delay shape high profile, then the probing step is not performed, and the admission request is denied [Page 873, Fig. 1]. If the delay shape profile is below the delay shape low profile after taking into account the new flow, the system automatically admits the new flow without probing the network [Page 873, Fig. 1].)

b. Deciding, based on said packet loss rate of previous transmissions, whether to determine a current packet loss rate by sending a burst of trial packets from a first node in a local area network to the local area network for which the packet loss rate of previous transmissions was determined, or to drop a connection attempt to that network (Pages 872-874, Particularly Sections 2-4) (See (a), Supra).

Qiu fails to disclose the method is for call admission control and tracks the packet loss rates of previous calls between local area networks such that the method further comprises determining a packet loss rate of previous calls to a local area network and deciding, based on

said packet loss rate of previous calls, whether to determine a current packet loss rate by sending a burst of trial packets from a first node in a local area network to the local area network for which the packet loss rate of previous calls was determined, or to drop a call attempt to that local area network. (i.e. The system of *Qiu* discloses the use of a generic admission control program that is not necessarily used for call admission control, although it appears that this use was contemplated by the authors and known in the art [See for Example Page 875, Reference [6], titled “Measurement-based call admission control…”]. *Qiu* further discloses that the system gathers the "previous" loss statistics used to determine if the network state is sufficiently “good” to warrant the sending of a probe to the network endpoint is based on prior active probing of delay and loss characteristics of the network, as opposed to passive observation of previous calls to the endpoint local area network. The system of *Komatsu* is provided to cure these deficiencies by showing that it was known to apply admission control to call admission control systems and to use the previous packet loss statistics from previous calls to determine the network state.) In the same field of endeavor, *Komatsu* discloses the method is for call admission control and tracks packet loss rates between local area networks such that the method comprises a method of call admission control for a continuous stream of data in packet switched networks including at least two local area networks that communicate with one another across a connecting network, the method comprising determining a packet loss rate of previous calls to a local area network, determining a current packet loss rate based on said packet loss rate of previous calls and deciding to drop a call attempt based on the current packet loss rate (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the

system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a call admission control scheme for tracking delay and packet loss between two LANs by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by modifying the system of *Qiu* to perform call, as opposed to session, based admission control, between to LANs as taught by *Komatsu* and to track the previous session statistics used by the system of *Qiu* to determine if the network condition is sufficiently good to warrant an active probe of the remote endpoint using delay and loss statistics collected from previous calls to remote networks, as opposed to previous probes to remote endpoints, as taught by *Komtsu*. The motive to combine is to allow the system of *Qui* to regulate call admission control using passive monitoring techniques that decrease the amount of probe traffic on the network, thereby increasing available network bandwidth.

Put another way, the system of *Qiu* discloses a base system in which the latency and loss characteristics from previous network sessions are used to determine if a probe is necessary to determine the current network state before admitting a network session and the system of *Komtsu* discloses a known improvement to network admission control involving using previous call latency and packet loss statistics in order to predict the current network state. Therefore, the claimed invention of using previous call latency and packet loss statistics to determine if a remote network is to be probed, as opposed to previous probe latency and loss

statistics, would have been obvious to a person of ordinary skill in the art at the time of the invention, in view the use of previous call latency and packet loss statistics to track network state without the need to consume network bandwidth by using active probes in the system of *Komatsu*.

Regarding claim 2, *Qiu* discloses a method further comprising the step of sending the trial packets, the method further comprising determining from said trial packets a current packet loss rate for sessions to the second network for which the packet loss rate of previous sessions was determined and deciding whether to drop the session attempt based at least in part on said current packet loss rate (Pages 872-873). (Whenever the system of *Qiu* detects an incoming connection control request, it compares the delay state from the last set of network probe packets with the delay state profiles [which are based on past packet loss rate and delay - i.e. the “packet loss rate of previous network transmissions”] [Pages 872-873, Section 2, Particularly Fig. 1]. If the delay shape is below the high delay shape profile but above the loss delay state profile, then the system will send a test probe comprising multiple packets [i.e. a “packet burst”] to the network to determine the current network loss characteristics [i.e. the “current packet loss rate”] and will admit or reject the transmission based on if the current loss rate is above or below and admission threshold [i.e. “decide to drop an attempt based on the current packet loss rate”] [Fig. 1, and Page 872, Last Paragraph, Carried on to Page 873][See also Page 873, Last Paragraph, Carried onto Page 874]. If the delay shape is above the delay shape high profile, then the probing step is not performed, and the admission request is denied [Page 873, Fig. 1]. If the delay shape profile is below the delay shape low profile after taking into account the new flow, the system automatically admits the new flow without probing the network [Page 873, Fig. 1].)

Qiu fails to disclose the method transmits packets between two local area networks or that the admission control technique is used to regulate call admission control such that the system further comprises determining from said trial packets a current packet loss rate for calls to the second local area network for which the packet loss rate of previous calls was determined and deciding whether to drop call attempt based at least in part on said current packet loss rate. In the same field of endeavor, *Komtsu* discloses the method transmits packets between two local area networks or that the admission control technique is used to regulate call admission control such that the system further comprises determining from said trial packets a current packet loss rate for calls to the second local area network for which the packet loss rate of previous calls was determined and deciding whether to drop call attempt based at least in part on said current packet loss rate (i.e. The system of *Qiu* discloses the use of a generic admission control program that is not necessarily used for call admission control, although it appears that this use was contemplated by the authors and known in the art [See for Example Page 875, Reference [6], titled “Measurement-based call admission control…”]. The system of *Komatsu* is provided to cure these deficiencies by showing that it was known to apply admission control to call admission control systems and to use the previous packet loss statistics from previous calls to determine the network state.) In the same field of endeavor, *Komatsu* discloses the method transmits packets between two local area networks and that the admission control technique is used to regulate call admission control such that the system further comprises determining from said trial packets a current packet loss rate for calls to the second local area network for which the packet loss rate of previous calls was determined and deciding whether to drop call attempt based at least in part on said current packet loss rate (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics

for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a call admission control scheme for tracking delay and packet loss between two LANs by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by modifying the system of *Qiu* to perform call, as opposed to session, based admission control and network probing. The motive to combine is to allow the system of *Qui* to regulate call admission control in a phone network by actively probing for the current network conditions, rather then relying on stale data.

Regarding claim 16, *Qiu* discloses the step of deciding to drop the call attempt based on the packet loss rate of previous calls to the network (Pages 872-874, Particularly Sections 2-4 - See Claim 1, Supra).

Qiu fails to disclose the calls are to the local area network. In the same field of endeavor, *Komatsu* discloses the calls are to the local area network (Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from

the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call [Column 3, Line 24]. The call is made between two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a call admission control scheme for tracking delay and packet loss between two LANs by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by modifying the system of *Qiu* to perform call, as opposed to session, based admission control and network probing of a remote local area network. The motive to combine is to allow the system of *Qui* to regulate call admission control between local area networks, allowing the use of cheap and widely available LAN VoIP devices.

5. **Claim 7** is rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1) as applied to claim 2 and further in view of and *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26).

Regarding claim 7, *Qiu* as modified by *Komatsu* fails to disclose the step of determining said current packet loss rate comprises transmitting the burst of trial packets to a node in the local area network for which the packet loss rate of previous calls was determined, reflecting the burst of trial packets received at that node back to the first node, and receiving the reflected burst of trial packets at the first node through the connecting network. In the same field of

endeavor, *Odam* discloses the step of determining said current packet loss rate comprises transmitting the burst of trial packets to a node in the local area network for which the packet loss rate of previous calls was determined, reflecting the burst of trial packets received at that node back to the first node, and receiving the reflected burst of trial packets at the first node through the connecting network (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data with packets having a priority and size corresponding to that of packets to be sent upon connection completion, and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two phones on different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more effective burst probing by transmitting a burst probe that more accurately reflects the packets to be transmitted on the admitted connection.

6. **Claims 8, 9 and 19** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1) and *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) as applied to claim 7 and further in view of *Oran*, et al. (US Pre Grant Publication No. 2006/0034188).

Regarding claim 8, *Qiu* as modified by *Komtasu* fails to disclose a method wherein the system comprises the use of VoIP Telephones connected to a LAN that utilizes a VoIP gateway for call admission control (i.e. In the system of *Komatsu* the only node on the local area network is the Exchange/IP packetizing unit [See Fig. 2, Element 24], as the actual telephone devices are traditional telephones and the Exchange/IP Packetizing unit acts as a conversion telephony gateway. The system of *Odom* is provided to cure this deficiency be showing the use of a telephony gateway that connects to a local area network that contains the endpoint VoIP telephones.) In the same field of endeavor, *Odom* discloses a method wherein the system comprises the use of VoIP Telephones connected to a LAN that utilizes a VoIP gateway for call admission control (Pages 1 and 4). (The system of *Odom* discloses the use of a telephony gateway that connect to a local area network containing VoIP Telephones [Page 4, Figure 4 and Fourth Paragraph].)

Therefore, since *Odom* discloses the use of VoIP endpoints connected to local area networks using a VoIP telephony gateway for admission control, it would have been obvious to combine the VoIP telephony devices of *Odom* with the system of *Qiu* as modified by *Komtasu* by replacing analog telephone devices with VoIP telephony devices and by replacing the call exchange/IP packetizing gateway with the VoIP gateway. The motive to combine is to provide a

system that operates completely using VoIP protocols, thereby eliminating the need to perform VoIP conversion at the gateway and allowing the use of modern VoIP telephones.

Qiu as modified by *Komtasu* and *Odom* fails to disclose a method wherein each said node comprises a telephone (i.e. The system of *Qiu* as modified by *Komtasu* and *Odom* discloses that the system may use VoIP telephony devices as the endpoints terminating calls, but it does not disclose that the system may test the call quality to a particular endpoint telephony device, as call quality is only monitored between the various VoIP gateways. The system of *Oran* is provided to cure this deficiency by showing it was known to measure call statistics to a remote telephone endpoint as opposed to a VoIP gateway associated with an endpoint's network.)

In the same field of endeavor, *Oran* discloses a method wherein the first node comprises a telephone a method wherein the each said node comprises a telephone (Paragraphs 0031-0038 and 0046). (The system of *Oran* discloses a system that simulates a voice call before initiation by sending a bi-directional stream of real time protocol (RTP) no-op packets between the sending and receiving VOIP telephones [Paragraphs 0031-0038]. The packets may be the same size as the actual media packets that are to follow [Paragraph 0046].)

Therefore, since *Oran* discloses call simulation between two telephone endpoints, it would have been obvious to combine the endpoint call simulation of *Oran* with the system of *Qiu* modified by *Komtasu* and *Odom* by having the telephone endpoints transmit a trial burst of data, as taught by *Oran* and reflecting the trial burst back to the sender, as taught by *Odom*. The motive to combine is to allow the telephone endpoints to test the connection using the endpoints as opposed to the gateway, thereby reducing the load on the gateway.

Regarding claim 9, *Qiu* fails to disclose said burst of trial packets comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is

completed. In the same field of endeavor, *Odom* discloses said burst of trial packets comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data with packets having a priority and size corresponding to that of packets to be sent upon connection completion, and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two phones on different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more effective burst probing by transmitting a burst probe that more accurately reflects the packets to be transmitted on the admitted connection.

Regarding claim 19, *Qiu* fails to disclose the burst of trial packets comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the call is completed. In the same field of endeavor, *Odam* discloses burst of trial packets comprises a plurality of packets having a size and priority that correspond to packets that are to be sent if the

call is completed (Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value). (The SAA protocol sends packets from the SAA client on the gateway device in the first LAN [Odom, Figure 4] to the server gateway in the other network/LAN. It then measures the packet loss rate of reflected packets to determine the packet loss rate of calls between the two networks [Odom, Page 19, SAA Protocol and Calculated Planned Impairment Value]. This value, along with others is used to perform client access control. The SAA protocol may also be configured to send probe packets based on the packet size of the codec to be used in the call using RTP headers to create a packet identical in size to one that would be used in a real voice conversation. The priority [i.e. IP precedence] of the packets may also be set to match the voice packets to be sent [Page 23, SAA Probe Format].)

Therefore, since *Odam* discloses using a reflected burst of trial data with packets having a priority and size corresponding to that of packets to be sent upon connection completion, and *Qiu* as modified by *Komatsu* discloses transmitting a burst between two phones on different local area networks, it would have been obvious to a person of ordinary skill at the time of the invention to replace the loss based burst probe of *Qiu* as modified by *Komatsu* with the burst probe of *Odam*. The motive to combine is to allow for more effective burst probing by transmitting a burst probe that more accurately reflects the packets to be transmitted on the admitted connection.

7. **Claims 14 and 17** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1) applied to claim 1 and

further in view of *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) and *Hosien*, et al. (US Patent No. 6,363,052 B1).

Regarding claim 14, *Qiu* fails to disclose the step of determining a packet loss rate of previous calls comprises the use of a gatekeeper comprising the first node to construct an estimate of attaining an acceptable call quality for the call attempt by using statistics about the quality of calls to a node in the local area network for which the packet loss rate of previous calls is being determined, said step of deciding whether to drop the call attempt being based at least in part on said estimate. In the same field of endeavor, *Komatsu* discloses the step of determining a packet loss rate of previous calls comprises the use of a gatekeeper comprising the first node to construct an estimate of attaining an acceptable call quality for the call attempt by using statistics about the quality of calls to a node in the local area network for which the packet loss rate of previous calls is being determined, said step of deciding whether to drop the call attempt being based at least in part on said estimate (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call or the call may be dropped automatically [Column 3, Line 24]. The call is made between endpoints in two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a call admission control scheme for tracking delay and packet loss between two LANs at a call gatekeeper by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by modifying the system of *Qiu* to track previous session statistics for gatekeepers associated with particular LANs in each gatekeeper device and to use the previous session statistics to estimate if the current conditions are sufficiently good to allow a call to proceed, proceed after a probe updates the current network conditions, or to be blocked without probing. The motive to combine is to allow the system of *Qui* to prevent the admission of excessive traffic to the network by blocking calls to the network when the estimated network condition is below an acceptable level.

Qiu as modified by *Komatsu* fails to disclose that the gatekeeper device is a gatekeeper for the local area network. In the same field of endeavor, *Odom* discloses the gatekeeper device is a gatekeeper for the local area network (Pages 1 and 4). (The system of *Odom* discloses the use of telephony gateways/gatekeepers that connect to a local area network containing VoIP telephones [Page 4, Figure 4 and Fourth Paragraph].)

Therefore, since *Odom* discloses the use of VoIP endpoints connected to local area networks using a VoIP telephony gateway for admission control, it would have been obvious to combine the VoIP telephony devices of *Odom* with the system of *Qiu* as modified by *Komtasu* by replacing analog telephone devices with VoIP telephony devices and by replacing the call exchange/IP packetizing gateway with the VoIP gateway. The motive to combine is to provide a system that operates completely using VoIP terminals, but still allows for centralized call admission control at a network gateway.

Qiu as modified by *Komatsu* and *Odam* fails to disclose the gatekeeper tracks the loss rate of previous calls and uses an estimate of the current loss rate for a call to be admitted based on the past call rate as a part of the decision to allow or block a call such that the method further comprises the gatekeeper uses an estimate of loss probability for the call attempt by using statistics about the success or failure of calls to a node in the local area network for which the packet loss rate of previous calls is being determined in said step of deciding whether to drop the call attempt being based at least in part on said estimate. In the same field of endeavor, *Hosien* discloses the gatekeeper tracks the loss rate of previous calls and uses an estimate of the current loss rate for a call to be admitted based on the past call rate as a part of the decision to allow or block a call such that the method further comprises the gatekeeper uses an estimate of loss probability for the call attempt by using statistics about the success or failure of calls to a node in the local area network for which the packet loss rate of previous calls is being determined in said step of deciding whether to drop the call attempt being based at least in part on said estimate (Figs. 2 and 3 and Column 3, Line 30 to 4, Lines 53). (The system of *Hosien* discloses that source switches/gateways connected to endpoint telephone devices [See Fig. 1, Elements 120 and 10] track the call completion rate [Fig. 3, Element 1020] [i.e. blocking probability] to remote switches/gateways that may also be connected to endpoint telephones [Fig. 1, Elements 170 and 20] [Column 3, Line 30 to 4, Lines 53]. The system then uses the congestion conditions of the remote endpoint combined with the measured call completion rate to estimate the rate at which it should reject calls to the remote endpoint [Column 4, Lines 36-46].)

Therefore, since *Hosien* suggests measuring the loss rate/probability of previous calls to a remote gateway and using the measured loss rate/probability along with other congestion metrics to determine if a call should be admitted or rejected, it would have been obvious to a

person of ordinary skill in the art at the time of the invention to combine the loss probabilities of *Hosien* with the system of *Qiu* as modified by *Komatsu* and *Odam* by tracking call loss rates/probabilities and using the call loss rate as a part of the admission criterion for call admission. The motive to combine is provided by *Hosien* and is to prevent overload of the destination switch/gateway (*Hosien*, Column 3, Lines 5-29).

Put another way, the system of *Qiu* as modified by *Komatsu* and *Odam* discloses a system that utilizes packet loss rates and latency to a particular VoIP gateway to determine if sufficient capacity exists on the network and at the endpoint to receive and process calls to a particular endpoint and then accepts or rejects incoming calls based on the availability of capacity. The system of *Hosien* discloses a known improvement for performing call admission control in which the call success rate/probability of previous calls to a particular gateway/gatekeeper connected to a telephone endpoint are used in conjunction with load metrics to determine if a call is to be admitted to the network. Therefore, the claimed invention of using measured call loss rates to an endpoint to predict the current call loss rate/probability and using the predicted call loss rate/probability as one of several admission control metrics was a part of the capabilities of a person of ordinary skill in the art at the time of the invention and would have been obvious to a person of ordinary skill in the art in view of the teaching of its use in other call admission control gateways to provide the benefit of preventing network overload at gateway network devices.

Regarding claim 17, *Qiu* fails to disclose the step of deciding whether to drop the call attempt comprises the use of said first node to make the decision. In the same field of endeavor, *Odam* discloses the step of deciding whether to drop the call attempt comprises use of said first node to make the decision (Page 4, Figure 4 and Fourth Paragraph). (The system of *Odam*

discloses that the VoIP gateway/first node is used to determine if a call is to be admitted to the network).

Therefore, since *Odam* discloses the use of the first node/gateway to perform admission control, it would have been obvious to a person of ordinary skill in the art a the time of the invention to use the gateway/gatekeeper of *Qui* as modified by *Komatsu* to perform admission control. The motive to combine is to perform call admission at the same location that collects call statistics, thereby simplifying the system by not requiring the transmission of call statistics to another node to make the admission decision.

8. **Claims 15 and 18** are rejected under 35 U.S.C. 103(a) as being unpatentable over *Qiu*, et al. (J. Qiu, H. Shao, W. Zhu and Y. Zhang, An End-to-End Probing Based Application Control Scheme for Multimedia Applications, IEEE International Conference on Multimedia, 2001, Pages 872-875) in view of *Komatsu*, et al. (US Patent No. 6, 914,900 B1) and *Odom* (Odom, Cisco VOIP Call Admission Control, August 2001, Pages 1-26) as applied to claim 7 and further in view of *Hosien*, et al. (US Patent No. 6,363,052 B1)

Regarding claim 15, *Qiu* fails to disclose the step of determining a packet loss rate of previous calls comprises the use of a gatekeeper comprising the first node to construct an estimate of attaining an acceptable call quality for the call attempt by using statistics about the quality of calls to a node in the local area network for which the packet loss rate of previous calls is being determined, said step of deciding whether to drop the call attempt being based at least in part on said estimate. In the same field of endeavor, *Komatsu* discloses the step of determining a packet loss rate of previous calls comprises the use of a gatekeeper comprising the first node to construct an estimate of attaining an acceptable call quality for the call attempt

by using statistics about the quality of calls to a node in the local area network for which the packet loss rate of previous calls is being determined, said step of deciding whether to drop the call attempt being based at least in part on said estimate (Column 5, Lines 5-20, Column 7, Lines 15-27, Column 3, Line 24 and Column 8, Lines 23-30). (The system of Komatsu discloses a system that obtains delay and packet loss data by tracking the delay and packet loss statistics for past calls to a network of interest [Column 5, Lines 5-20, Column 6, Line 65 to Column 7, Line 6 and Column 7, Lines 15-27]. When a call is made, the system references the loss statistics for the last call to that IP endpoint at the same time and date from the current endpoint and if the loss rate is acceptable, the call is admitted. If the loss rate is unacceptable, the user is notified and may drop the call or the call may be dropped automatically [Column 3, Line 24]. The call is made between endpoints in two different LANs, as indicated by the fact that traffic form the IP packetizing unit passes through a router to reach its destination [Column 8, Lines 23-30].)

Therefore, since *Komatsu* discloses a call admission control scheme for tracking delay and packet loss between two LANs at a call gatekeeper by monitoring the delay and loss statistics of calls between the networks, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the call based network tracking and admission control of *Komatsu* with the system of *Qiu* by modifying the system of *Qiu* to track previous session statistics for gatekeepers associated with particular LANs in each gatekeeper device and to use the previous session statistics to estimate if the current conditions are sufficiently good to allow a call to proceed, proceed after a probe updates the current network conditions, or to be blocked without probing. The motive to combine is to allow the system of *Qui* to prevent the admission of excessive traffic to the network by blocking calls to the network when the estimated network condition is below an acceptable level.

Qiu as modified by *Komatsu* fails to disclose that the gatekeeper device is a gatekeeper for the local area network. In the same field of endeavor, *Odom* discloses the gatekeeper device is a gatekeeper for the local area network (Pages 1 and 4). (The system of *Odom* discloses the use of telephony gateways/gatekeepers that connect to a local area network containing VoIP telephones [Page 4, Figure 4 and Fourth Paragraph].)

Therefore, since *Odom* discloses the use of VoIP endpoints connected to local area networks using a VoIP telephony gateway for admission control, it would have been obvious to combine the VoIP telephony devices of *Odom* with the system of *Qiu* as modified by *Komtasu* by replacing analog telephone devices with VoIP telephony devices and by replacing the call exchange/IP packetizing gateway with the VoIP gateway. The motive to combine is to provide a system that operates completely using VoIP terminals, but still allows for centralized call admission control at a network gateway.

Qiu as modified by *Komatsu* and *Odam* fails to disclose the gatekeeper tracks the loss rate of previous calls and uses an estimate of the current loss rate for a call to be admitted based on the past call rate as a part of the decision to allow or block a call such that the method further comprises the gatekeeper uses an estimate of loss probability for the call attempt by using statistics about the success or failure of calls to a node in the local area network for which the packet loss rate of previous calls is being determined in said step of deciding whether to drop the call attempt being based at least in part on said estimate. In the same field of endeavor, *Hosien* discloses the gatekeeper tracks the loss rate of previous calls and uses an estimate of the current loss rate for a call to be admitted based on the past call rate as a part of the decision to allow or block a call such that the method further comprises the gatekeeper uses an estimate of loss probability for the call attempt by using statistics about the success or failure of calls to a node in the local area network for which the packet loss rate of previous calls is

being determined in said step of deciding whether to drop the call attempt being based at least in part on said estimate (Figs. 2 and 3 and Column 3, Line 30 to 4, Lines 53). (The system of *Hosien* discloses that source switches/gateways connected to endpoint telephone devices [See Fig. 1, Elements 120 and 10] track the call completion rate [Fig. 3, Element 1020] [i.e. blocking probability] to remote switches/gateways that may also be connected to endpoint telephones [Fig. 1, Elements 170 and 20] [Column 3, Line 30 to 4, Lines 53]. The system then uses the congestion conditions of the remote endpoint combined with the measured call completion rate to estimate the rate at which it should reject calls to the remote endpoint [Column 4, Lines 36-46].)

Therefore, since *Hosien* suggests measuring the loss rate/probability of previous calls to a remote gateway and using the measured loss rate/probability along with other congestion metrics to determine if a call should be admitted or rejected, it would have been obvious to a person of ordinary skill in the art at the time of the invention to combine the loss probabilities of *Hosien* with the system of *Qiu* as modified by *Komatsu* and *Odam* by tracking call loss rates/probabilities and using the call loss rate as a part of the admission criterion for call admission. The motive to combine is provided by *Hosien* and is to prevent overload of the destination switch/gateway (*Hosien*, Column 3, Lines 5-29).

Put another way, the system of *Qiu* as modified by *Komatsu* and *Odam* discloses a system that utilizes packet loss rates and latency to a particular VoIP gateway to determine if sufficient capacity exists on the network and at the endpoint to receive and process calls to a particular endpoint and then accepts or rejects incoming calls based capacity availability. The system of *Hosien* discloses a known improvement for performing call admission control in which the call success rate/probability of previous calls to a particular gateway/gatekeeper connected to a telephone endpoint are used in conjunction with load metrics to determine if a call is to be

admitted to the network. Therefore, the claimed invention of using measured call loss rates to an endpoint to predict the current call loss rate/probability and using the predicted call loss rate/probability as one of several admission control metrics was a part of the capabilities of a person of ordinary skill in the art at the time of the invention and would have been obvious to a person of ordinary skill in the art in view of the teaching of its use in other call admission control gateways to provide the benefit of preventing network overload at core and gateway network devices.

Regarding claim 18, *Qiu* fails to disclose the step of deciding whether to drop the call attempt comprises use of said first node to make the decision. In the same field of endeavor, *Odam* discloses the step of deciding whether to drop the call attempt comprises use of said first node to make the decision (Page 4, Figure 4 and Fourth Paragraph). (The system of *Odam* discloses that the VoIP gateway/first node is used to determine if a call is to be admitted to the network).

Therefore, since *Odam* discloses the use of the first node/gateway to perform admission control, it would have been obvious to a person of ordinary skill in the art a the time of the invention to use the gateway/gatekeeper of *Qui* as modified by *Komatsu* to perform admission control. The motive to combine is to perform call admission at the same location that collects call statistics, thereby simplifying the system by not requiring the transmission of call statistics to another node to make the admission decision.

Response to Arguments

9. Most of Applicant's Arguments filed 14 February 2011 have been fully considered but they are not persuasive.

With regard to Applicant's Arguments that the system of *Qiu* does not use previous packet loss rates to determine if it is to send a trial burst of packets to a remote network, Applicants Arguments have been considered and are not persuasive (See Applicant's Arguments and Remarks, Pages 8-9).

The issue at hand is does the system of *Qiu* teach the use of the packet loss rate of previous transmissions to determine if it is to send a trial burst to a remote network or drop a call attempt? This is answered in the affirmative, as the system of *Qiu* uses a set of delay state graphs based on the packet loss rate of previously transmitted probes to determine if it is to probe the remote network before call admission or outright drop a call attempt.

The system of *Qiu* discloses a call admission algorithm that has three possible states for call admission. In the first state, the current delay shape of admitted packets is greater then the delay shape high, and all calls are dropped (Page 873, Right Column, Section 4). In the second state the current delay state is greater then the delay shape low but less then the delay state high, and the remote endpoint is probed using a burst of trial data to determine if the call should be admitted (Page 873, Right Column, Section 4 to Page 874, Left Column, Section 4). In the third state, the delay shape is less then the delay shape low and all calls are admitted. Therefore, as an initial matter, *Qiu* discloses the existence of three states in which the call is unconditionally dropped, admitted only after a test probe and unconditionally accepted, respectively.

Therefore, the issue remaining is does the determination of the existence of the first and second state, in which a call is respectively unconditionally dropped and a trial burst is sent to a remote network, depend on the packet loss rate of previous transmissions? Looking to section 3 of *Qiu* it can be seen that the delay shape high and the delay shape low thresholds are calculated directly from the two delay state graphs and that the contents of the delay state graphs are based on the delays above and below a desirable packet loss rate [i.e. state 0 is the delay when the loss rate is <1% and state 1 is the delay when the loss rate is >1%] [Page 873, Section 3]. Therefore, the determination between the first and second state is based on the loss rate of previous transmissions to the network.

Applicant's Argument that *Qiu* fails to disclose the claimed invention, as it is directed only to the operation of lightly congested networks and only considers delay have been considered and are not persuasive (See Applicant's Arguments and Remarks, Page 9). As an initial matter, it is clear that the algorithm of *Qiu* was intended to be used in both lightly and heavily congested networks (See, For Example, The Simulations on Page 874 showing that bottleneck links are saturated to nearly 90% utilization). Furthermore, although Applicant's Arguments correctly point out that the system of *Qiu* uses delay as a predictor of network congestion, this does not diminish the fact that the packet loss rates are also considered (See, For Example, Page 872, "To address this issue, we use both packet loss rate and delay as the criteria in admission control.")

Applicant's Argument that *Qiu* uses only the current delay to determine if a call is to be admitted/dropped or an endpoint probed have been considered and are not persuasive, as the admission decision is based both on the current delay and on delay thresholds calculated from historic packet loss rates (See Applicant's Arguments and Remarks, Pages 9-10). Therefore, previous packet loss rates are used to determine if a call is dropped or an endpoint probed.

Applicant's Argument that *Komatsu* fails to disclose the use of call because it is directed to "call routing" as opposed to call admission have been considered and are not persuasive (See Applicant's Arguments and Remarks, Pages 10-11). *Komatsu* teaches a technique for determining if a call is to be admitted to an IP network. If, based on present and historic call statistics, the IP network endpoint of the call is deemed suitable, then the call is admitted to the IP network, if not it is rerouted to a telephony network. Therefore, the system of *Komatsu* discloses the use of call admission control. Furthermore, even if this were not the case, the primary reference of *Qiu* discloses pure network admission control. Finally, even if both *Qiu* and *Komatsu* failed to disclose call admission control the system of *Odom* discloses these features and could be combined with the system of *Qiu* as modified by *Komatsu* as outlined with respect to claims 8 and 9, supra.

Applicant's Argument that *Komatsu* fails to meet the requirements of the claims because "Although it might be using an IP network as its trunk network, all that is happening is that the traditional trunk connection has been replaced" (See Applicant's Arguments and Remarks, Pages 10-12), Applicant's Arguments have been considered and are not persuasive, as the system of *Komatsu* meets all the claimed limitations, which do not require that the actual telephony endpoints be VoIP devices connected to a LAN, but only requires that the actual telephony gateway be connected to a LAN, a requirement met by the gateway of *Komatsu*, which is connected to an IP LAN as describe dwith respect to claim 1, supra. Furthermore, even if *Komatsu* failed to disclose call admission control the system of *Odom* discloses these features and could be combined with the system of *Qiu* as modified by *Komatsu* as outlined with respect to claims 8 and 9, supra.

10. The remainder of Applicant's arguments with respect to claims 1, 2, 7-9 and 14-19 have been considered but are moot in view of the new ground(s) of rejection.

Conclusion

11. Applicant's amendment necessitated the new ground(s) of rejection presented in this Office action. Accordingly, **THIS ACTION IS MADE FINAL**. See MPEP § 706.07(a). Applicant is reminded of the extension of time policy as set forth in 37 CFR 1.136(a).

A shortened statutory period for reply to this final action is set to expire THREE MONTHS from the mailing date of this action. In the event a first reply is filed within TWO MONTHS of the mailing date of this final action and the advisory action is not mailed until after the end of the THREE-MONTH shortened statutory period, then the shortened statutory period will expire on the date the advisory action is mailed, and any extension fee pursuant to 37 CFR 1.136(a) will be calculated from the mailing date of the advisory action. In no event, however, will the statutory period for reply expire later than SIX MONTHS from the date of this final action.

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Christopher Crutchfield whose telephone number is (571) 270-3989. The examiner can normally be reached on Monday through Friday 8:00 AM to 5:00 PM EST.

If attempts to reach the examiner by telephone are unsuccessful, the examiner’s supervisor, Daniel Ryman can be reached on (571) 272-3152. The fax phone number for the organization where this application or proceeding is assigned is 571-273-8300.

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free). If you would like assistance from a USPTO Customer Service Representative or access to the automated information system, call 800-786-9199 (IN USA OR CANADA) or 571-272-1000.

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Examiner, Art Unit 2466
4/22/2011

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